

Microphones

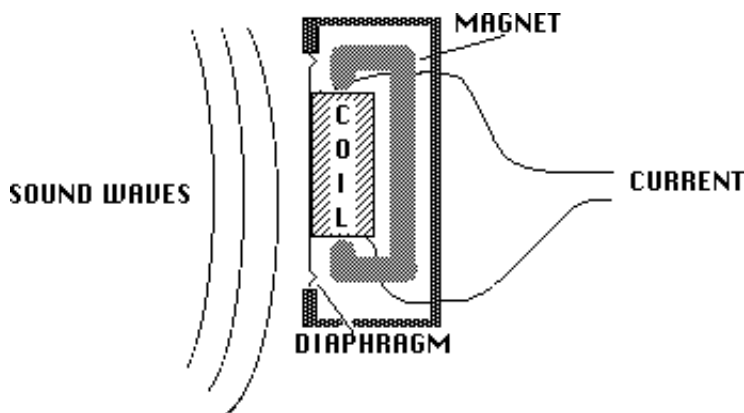
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I. How They Work.

A microphone is an example of a transducer, a device that changes information from one form to another. Sound information exists as patterns of air pressure; the microphone changes this information into patterns of electric current. The recording engineer is interested in the accuracy of this transformation, a concept he thinks of as fidelity.

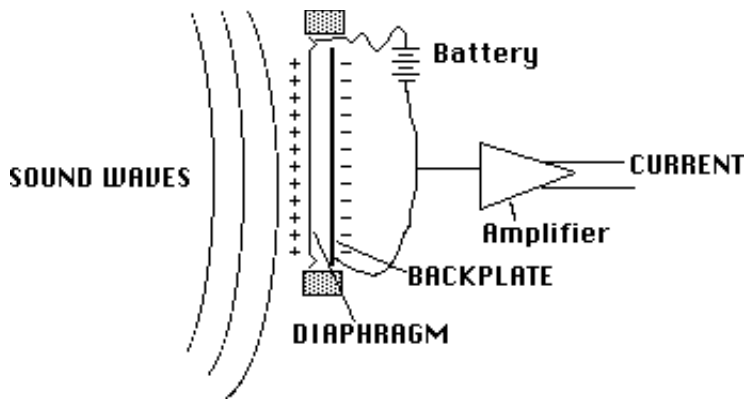
A variety of mechanical techniques can be used in building microphones. The two most commonly encountered in recording studios are the magneto-dynamic and the variable condenser designs.

THE DYNAMIC MICROPHONE.



In the magneto-dynamic, commonly called dynamic, microphone, sound waves cause movement of a thin metallic diaphragm and an attached coil of wire. A magnet produces a magnetic field which surrounds the coil, and motion of the coil within this field causes current to flow. The principles are the same as those that produce electricity at the utility company, realized in a pocket-sized scale. It is important to remember that current is produced by the motion of the diaphragm, and that the amount of current is determined by the speed of that motion. This kind of microphone is known as **velocity sensitive**.

THE CONDENSER MICROPHONE.



In a condenser microphone, the diaphragm is mounted close to, but not touching, a rigid backplate. (The plate may or may not have holes in it.) A battery is connected to both pieces of metal, which produces an electrical potential, or charge, between them. The amount of charge is determined by the voltage of the battery, the area of the diaphragm and backplate, and the distance between the two. This distance changes as the diaphragm moves in response to sound. When the distance changes, current flows in the wire as the battery maintains the correct charge. The amount of current is essentially proportional to the **displacement** of the diaphragm, and is so small that it must be electrically amplified before it leaves the microphone.

A common variant of this design uses a material with a permanently imprinted charge for the diaphragm. Such a material is called an **electret** and is usually a kind of plastic. (You often get a piece of plastic with a permanent charge on it when you unwrap a record. Most plastics conduct electricity when they are hot but are insulators when they cool.) Plastic is a pretty good material for making diaphragms since it can be dependably produced to fairly exact specifications. (Some popular dynamic microphones use plastic diaphragms.) The major disadvantage of electrets is that they lose their charge after a few years and cease to work.

II. Specifications

There is no inherent advantage in fidelity of one type of microphone over another. Condenser types require batteries or power from the mixing console to operate, which is occasionally a hassle, and dynamics require shielding from stray magnetic fields, which makes them a bit heavy sometimes, but very fine microphones are available of both styles. The most important factor in choosing a microphone is how it sounds in the required application. The following issues must be considered:

Sensitivity.

This is a measure of how much electrical output is produced by a given sound. This is a vital specification if you are trying to record very tiny sounds, such as a turtle snapping its jaw, but should be considered in any situation. If you put an insensitive mic on a quiet instrument, such as an acoustic guitar, you will have to increase the gain of the mixing console, adding noise to the mix. On the other hand, a very sensitive mic on vocals might overload the input electronics of the mixer or tape deck, producing distortion.

Overload characteristics.

Any microphone will produce distortion when it is overdriven by loud sounds. This is caused by various factors. With a dynamic, the coil may be pulled out of the magnetic field; in a condenser, the internal amplifier might clip. Sustained overdriving or extremely loud sounds can permanently distort the diaphragm, degrading performance at ordinary sound levels. Loud sounds are encountered more often than you might think, especially if you place the mic very close to instruments. (Would you put your ear in the bell of a trumpet?) You usually get a choice between high sensitivity and high overload points, although occasionally there is a switch on the microphone for different situations.

Linearity, or Distortion.

This is the feature that runs up the price of microphones. The distortion characteristics of a mic are determined mostly by the care with which the diaphragm is made and mounted. High volume production methods can turn out an adequate microphone, but the distortion performance will be a matter of luck. Many manufacturers have several model numbers for what is essentially the same device. They build a batch, and then test the mics and charge a premium price for the good ones. The really big names throw away mic capsules that don't meet their standards. (If you buy one Neumann mic, you are paying for five!)

No mic is perfectly linear; the best you can do is find one with distortion that complements the sound you are trying to record. This is one of the factors of the microphone mystique discussed later.

Frequency response.

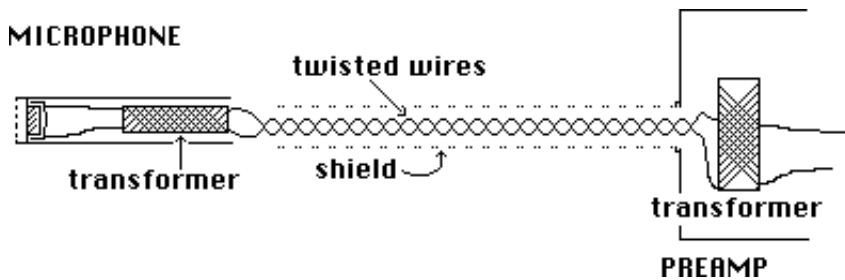
A flat frequency response has been the main goal of microphone companies for the last three or four decades. In the fifties, mics were so bad that console manufacturers began adding equalizers to each input to compensate. This effort has now paid off to the point where most professional microphones are respectably flat, at least for sounds originating in front. The major exceptions are mics with deliberate emphasis at certain frequencies that are useful for some applications. This is another part of the microphone mystique. Problems in frequency response are mostly encountered with sounds originating behind the mic, as discussed in the next section.

Noise.

Microphones produce a very small amount of current, which makes sense when you consider just how light the moving parts must be to accurately follow sound waves. To be useful for recording or other electronic processes, the signal must be amplified by a factor of over a thousand. Any electrical noise produced by the microphone will also be amplified, so even slight amounts are intolerable. Dynamic microphones are essentially noise free, but the electronic circuit built into condenser types is a potential source of trouble, and must be carefully designed and constructed of premium parts.

Noise also includes unwanted pickup of mechanical vibration through the body of the microphone. Very sensitive designs require elastic shock mountings, and mics intended to be held in the hand need to have such mountings built inside the shell.

The most common source of noise associated with microphones is the wire connecting the mic to the console or tape deck. A mic preamp is very similar to a radio receiver, so the cable must be prevented from becoming an antenna. The basic technique is to surround the wires that carry the current to and from the mic with a flexible metallic shield, which deflects most radio energy. A second technique, which is more effective for the low frequency hum induced by the power company into our environment, is to balance the line:



Current produced by the microphone will flow down one wire of the twisted pair, and back along the other one. Any current induced in the cable from an outside source would tend to flow the same way in both wires, and such currents cancel each other in the transformers. This system is expensive.

Microphone Levels

As I said, microphone outputs are of necessity very weak signals, generally around -60dBm. (The specification is the power produced by a sound pressure of 10 uBar) The output impedance will depend on whether the mic has a transformer balanced output. If it does not, the microphone will be labeled "high impedance" or "hi Z" and must be connected to an appropriate input. The cable used must be kept short, less than 10 feet or so, to avoid noise problems.

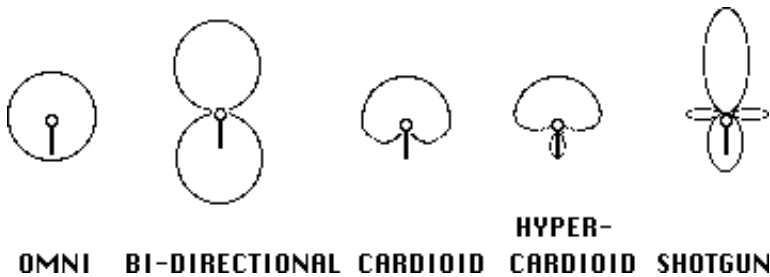
If a microphone has a transformer, it will be labeled low impedance, and will work best with a balanced input mic preamp. The cable can be several hundred feet long with no problem. Balanced output, low impedance microphones are expensive, and generally found in professional applications. Balanced outputs must have three pin connectors ("Cannon plugs"), but not all mics with those plugs are really balanced. Microphones with standard or miniature phone plugs are high impedance. A balanced mic can be used with a high impedance input with a suitable adapter.

You can see from the balanced connection diagram that there is a transformer at the input of the console preamp. (Or, in lieu of a transformer, a complex circuit to do the same thing.) This is the most significant difference between professional preamplifiers and the type usually found on home tape decks. You can buy transformers that are designed to add this feature to a consumer deck for about \$20 each. (Make sure you are getting a transformer and not just an adapter for the connectors.) With these accessories you can use professional quality microphones, run cables over a hundred feet with no hum, and because the transformers

boost the signal somewhat, make recordings with less noise. This will not work with a few inexpensive cassette recorders, because the strong signal causes distortion. Such a deck will have other problems, so there is little point trying to make a high fidelity recording with it anyway.

III. Pick Up Patterns

Many people have the misconception that microphones only pick up sound from sources they are pointed at, much as a camera only photographs what is in front of the lens. This would be a nice feature if we could get it, but the truth is we can only approximate that action, and at the expense of other desirable qualities.



MICROPHONE PATTERNS

These are polar graphs of the output produced vs. the angle of the sound source. The output is represented by the radius of the curve at the incident angle.

Omni

The simplest mic design will pick up all sound, regardless of its point of origin, and is thus known as an omnidirectional microphone. They are very easy to use and generally have good to outstanding frequency response. To see how these patterns are produced, here's a sidebar on [directional microphones](#).

Bi-directional

It is not very difficult to produce a pickup pattern that accepts sound striking the front or rear of the diaphragm, but does not respond to sound from the sides. This is the way any diaphragm will behave if sound can strike the front and back equally. The rejection of undesired sound is the best achievable with any design, but the fact that the mic accepts sound from both ends makes it difficult to use in many situations. Most often it is placed above an instrument. Frequency response is just as good as an omni, at least for sounds that are not too close to the microphone.

Cardioid

This pattern is popular for sound reinforcement or recording concerts where audience noise is a possible problem. The concept is great, a mic that picks up sounds it is pointed at. The reality is different. The first problem is that sounds from the back are not completely rejected, but merely reduced about 10-30 dB. This can surprise careless users. The second problem, and a severe one, is that the actual shape of the pickup

pattern varies with frequency. For low frequencies, this is an omnidirectional microphone. A mic that is directional in the range of bass instruments will be fairly large and expensive. Furthermore, the frequency response for signals arriving from the back and sides will be uneven; this adds an undesired coloration to instruments at the edge of a large ensemble, or to the reverberation of the concert hall.

A third effect, which may be a problem or may be a desired feature, is that the microphone will emphasize the low frequency components of any source that is very close to the diaphragm. This is known as the "[proximity effect](#)", and many singers and radio announcers rely on it to add "chest" to a basically light voice. Close, in this context, is related to the size of the microphone, so the nice large mics with even back and side frequency response exhibit the strongest presence effect. Most cardioid mics have a built in lowcut filter switch to compensate for proximity. Missetting that switch can cause hilarious results. Bidirectional mics also exhibit this phenomenon.

Tighter Patterns

It is possible to exaggerate the directionality of cardioid type microphones, if you don't mind exaggerating some of the problems. The Hypercardioid pattern is very popular, as it gives a better overall rejection and flatter frequency response at the cost of a small back pickup lobe. This is often seen as a good compromise between the cardioid and bidirectional patterns. A "shotgun" mic carries these techniques to extremes by mounting the diaphragm in the middle of a pipe. The shotgun is extremely sensitive along the main axis, but possesses pronounced extra lobes which vary drastically with frequency. In fact, the frequency response of this mic is so bad it is usually electronically restricted to the voice range, where it is used to record dialogue for film and video.

Stereo microphones

You don't need a special microphone to record in stereo, you just need two (see below). A so called stereo microphone is really two microphones in the same case. There are two kinds: extremely expensive professional models with precision matched capsules, adjustable capsule angles, and remote switching of pickup patterns; and very cheap units (often with the capsules oriented at 180 deg.) that can be sold for high prices because they have the word stereo written on them.

IV. Typical Placement

Single microphone use

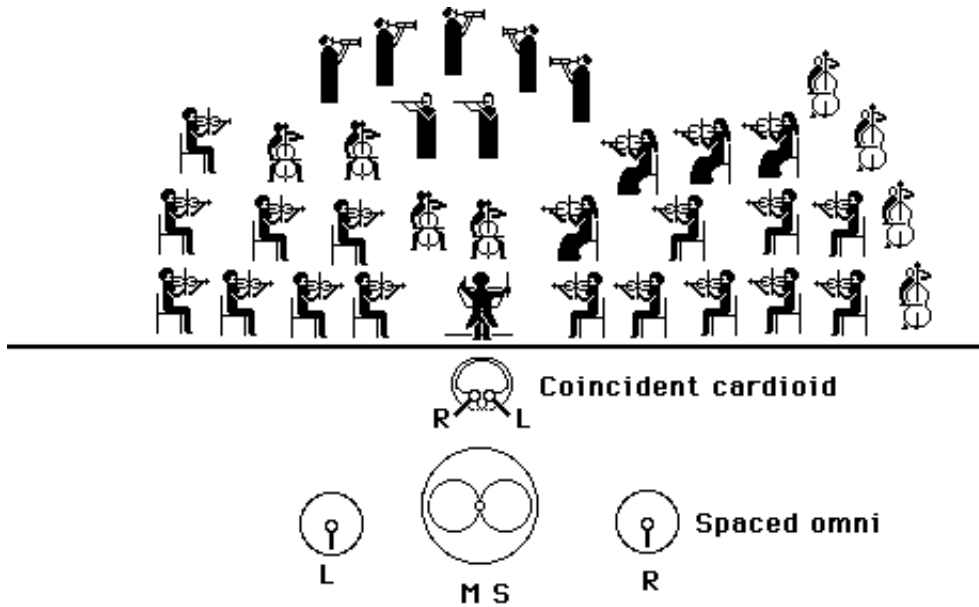
Use of a single microphone is pretty straightforward. Having chosen one with appropriate sensitivity and pattern, (and the best distortion, frequency response, and noise characteristics you can afford), you simply mount it where the sounds are. The practical range of distance between the instrument and the microphone is determined by the point where the sound overloads the microphone or console at the near end, and the point where ambient noise becomes objectionable at the far end. Between those extremes it is largely a matter of taste and experimentation.

If you place the microphone close to the instrument, and listen to the results, you will find the location of the mic affects the way the instrument sounds on the recording. The timbre may be odd, or some notes may be louder than others. That is because the various components of an instrument's sound often come from different parts of the instrument body (the highest note of a piano is nearly five feet from the lowest), and we are used to hearing an evenly blended tone. A close in microphone will respond to some locations on the instrument more than others because the difference in distance from each to the mic is proportionally large. A good rule of thumb is that the blend zone starts at a distance of about twice the length of the instrument. If you are recording several instruments, the distance between the players must be treated the same way.

If you place the microphone far away from the instrument, it will sound as if it is far away from the instrument. We judge sonic distance by the ratio of the strength of the direct sound from the instrument (which is always heard first) to the strength of the reverberation from the walls of the room. When we are physically present at a concert, we use many cues beside the sounds to keep our attention focused on the performance, and we are able to ignore any distractions there may be. When we listen to a recording, we don't have those visual clues to what is happening, and find anything extraneous that is very audible annoying. For this reason, the best seat in the house is not a good place to record a concert. On the other hand, we do need some reverberation to appreciate certain features of the music. (That is why some types of music sound best in a stone church) Close microphone placement prevents this. Some engineers prefer to use close miking techniques to keep noise down and add artificial reverberation to the recording, others solve the problem by mounting the mic very high, away from audience noise but where adequate reverberation can be found.

Stereo

Stereo sound is an illusion of spaciousness produced by playing a recording back through two speakers. The success of this illusion is referred to as the image. A good image is one in which each instrument is a natural size, has a distinct location within the sound space, and does not move around. The main factors that establish the image are the relative strength of an instrument's sound in each speaker, and the timing of arrival of the sounds at the listener's ear. In a studio recording, the stereo image is produced artificially. Each instrument has its own microphone, and the various signals are balanced in the console as the producer desires. In a concert recording, where the point is to document reality, and where individual microphones would be awkward at best, it is most common to use two mics, one for each speaker.



Microphone placement for stereo recording.

Spaced microphones

The simplest approach is to assume that the speakers will be eight to ten feet apart, and place two microphones eight to ten feet apart to match. Either omnis or cardioids will work. When played back, the results will be satisfactory with most speaker arrangements. (I often laugh when I attend concerts and watch people using this setup fuss endlessly with the precise placement of the mics. This technique is so forgiving that none of their efforts will make any practical difference.)

The big disadvantage of this technique is that the mics must be rather far back from the ensemble- at least as far as the distance from the leftmost performer to the rightmost. Otherwise, those instruments closest to the microphones will be too prominent. There is usually not enough room between stage and audience to achieve this with a large ensemble, unless you can suspend the mics or have two very tall stands.

Coincident cardioids

There is another disadvantage to the spaced technique that appears if the two channels are ever mixed together into a monophonic signal. (Or broadcast over the radio, for similar reasons.) Because there is a large distance between the mics, it is quite possible that sound from a particular instrument would reach each mic at slightly different times. (Sound takes 1 millisecond to travel a foot.) This effect creates phase differences between the two channels, which results in severe frequency response problems when the signals are combined. You seldom actually lose notes from this interference, but the result is an uneven, almost shimmering sound. The various coincident techniques avoid this problem by mounting both mics in almost the same spot.

This is most often done with two cardioid microphones, one pointing slightly left, one slightly right. The microphones are often pointing toward each other, as this places the diaphragms within a couple of inches of each other, totally eliminating phase problems. No matter how they are mounted, the microphone that points

to the left provides the left channel. The stereo effect comes from the fact that the instruments on the right side are on-axis for the right channel microphone and somewhat off-axis (and therefore reduced in level) for the other one. The angle between the microphones is critical, depending on the actual pickup pattern of the microphone. If the mics are too parallel, there will be little stereo effect. If the angle is too wide, instruments in the middle of the stage will sound weak, producing a hole in the middle of the image. [Incidentally, to use this technique, you must know which way the capsule actually points. There are some very fine German cardioid microphones in which the diaphragm is mounted so that the pickup is from the side, even though the case is shaped just like many popular end addressed models. (The front of the mic in question is marked by the trademark medallion.) I have heard the results where an engineer mounted a pair of these as if the axis were at the end. You could hear one cello player and the tympani, but not much else.]

You may place the microphones fairly close to the instruments when you use this technique. The problem of balance between near and far instruments is solved by aiming the mics toward the back row of the ensemble; the front instruments are therefore off axis and record at a lower level. You will notice that the height of the microphones becomes a critical adjustment.

M.S.

The most elegant approach to coincident miking is the M.S. or middle-side technique. This is usually done with a stereo microphone in which one element is omnidirectional, and the other bidirectional. The bidirectional element is oriented with the axis running parallel to the stage, rejecting sound from the center. The omni element, of course, picks up everything. To understand the next part, consider what happens as instrument is moved on the stage. If the instrument is on the left half of the stage, a sound would first move the diaphragm of the bidirectional mic to the right, causing a positive voltage at the output. If the instrument is moved to center stage, the microphone will not produce any signal at all. If the instrument is moved to the right side, the sound would first move the diaphragm to the left, producing a negative voltage. You can then say that instruments on one side of the stage are 180 degrees out of phase with those on the other side, and the closer they are to the center, the weaker the signal produced.

Now the signals from the two microphones are not merely kept in two channels and played back over individual speakers. The signals are combined in a circuit that has two outputs; for the left channel output, the bidirectional output is added to the omni signal. For the right channel output, the bidirectional output is subtracted from the omni signal. This gives stereo, because an instrument on the right produces a negative signal in the bidirectional mic, which when added to the omni signal, tends to remove that instrument, but when subtracted, increases the strength of the instrument. An instrument on the left suffers the opposite fate, but instruments in the center are not affected, because their sound does not turn up in the bidirectional signal at all.

M.S. produces a very smooth and accurate image, and is entirely mono compatible. The only reason it is not used more extensively is the cost of the special microphone and decoding circuit, well over \$1,000.

Large ensembles

The above techniques work well for concert recordings in good halls with small ensembles. When recording large groups in difficult places, you will often see a combination of spaced and coincident pairs. This does produce a kind of chorusing when the signals are mixed, but it is an attractive effect and not very different from the sound of string or choral ensembles any way. When balance between large sections and soloists cannot be achieved with the basic setup, extra microphones are added to highlight the weaker instruments. A very common problem with large halls is that the reverberation from the back seems late when compared to the direct sound taken at the edge of the stage. This can be helped by placing a mic at the rear of the audience area to get the ambient sound into the recording sooner.

Studio techniques

A complete description of all of the procedures and tricks encountered in the recording studio would fill several books. These are just a few things you might see if you dropped in on the middle of a session.

Individual mics on each instrument.

This provides the engineer with the ability to adjust the balance of the instruments at the console, or, with a multitrack recorder, after the musicians have gone home. There may be eight or nine mics on the drum set alone.

Close mic placement.

The microphones will usually be placed rather close to the instruments. This is partially to avoid problems that occur when an instrument is picked up in two non-coincident mics, and partially to modify the sound of the instruments (to get a "honky-tonk" effect from a grand piano, for instance).

Acoustic fences around instruments, or instruments in separate rooms.

The interference that occurs when an instrument is picked up by two mics that are mixed is a very serious problem. You will often see extreme measures, such as a bass drum stuffed with blankets to muffle the sound, and then electronically processed to make it sound like a drum again.

Everyone wearing headphones.

Studio musicians often play to "click tracks", which are not recorded metronomes, but someone tapping the beat with sticks and occasionally counting through tempo changes. This is done when the music must be synchronized to a film or video, but is often required when the performer cannot hear the other musicians because of the isolation measures described above.

20 or 30 takes on one song.

Recordings require a level of perfection in intonation and rhythm that is much higher than that acceptable in concert. The finished product is usually a composite of several takes.

Pop filters in front of mics.

Some microphones are very sensitive to minor gusts of wind--so sensitive in fact that they will produce a loud pop if you breath on them. To protect these mics (some of which can actually be damaged by blowing in them) engineers will often mount a nylon screen between the mic and the artist. This is not the most common reason for using pop filters though:

Vocalists like to move around when they sing; in particular, they will lean into microphones. If the singer is very close to the mic, any motion will produce drastic changes in level and sound quality. (You have seen this with inexperienced entertainers using hand held mics.) Many engineers use pop filters to keep the artist at the proper distance. The performer may move slightly in relation to the screen, but that is a small proportion of the distance to the microphone.

V. The Microphone Mystique

There is an aura of mystery about microphones. To the general public, a recording engineer is something of a magician, privy to a secret arcana, and capable of supernatural feats. A few modern day engineers encourage this attitude, but it is mostly a holdover from the days when studio microphones were expensive and fragile, and most people never dealt with any electronics more complex than a table radio. There are no secrets to recording; the art is mostly a commonsense application of the principles already discussed in this paper. If there is an arcana, it is an accumulation of trivia achieved through experience with the following problems:

Matching the microphone to the instrument.

There is no wrong microphone for any instrument. Every engineer has preferences, usually based on mics with which he is familiar. Each mic has a unique sound, but the differences between good examples of any one type are pretty minor. The artist has a conception of the sound of his instrument, (which may not be accurate) and wants to hear that sound through the speakers. Frequency response and placement of the microphone will affect that sound; sometimes you need to exaggerate the features of the sound the client is looking for.

Listening the proper way.

It is easy to forget that the recording engineer is an illusionist- the result will never be confused with reality by the listener. Listeners are in fact very forgiving about some things. It is important that the engineer be able to focus his attention on the main issues and not waste time with interesting but minor technicalities. It is important that the engineer know what the main issues are. An example is the noise/distortion tradeoff. Most listeners are willing to ignore a small amount of distortion on loud passages (in fact, they expect it), but would be annoyed by the extra noise that would result if the engineer turned the recording level down to avoid it. One technique for encouraging this attention is to listen to recordings over a variety of sound systems, good and bad.

Learning for yourself.

Many students come to me asking for a book or a course of study that will easily make them a member of this elite company. There are books, and some schools have courses in recording, but they do not supply the essential quality the professional recording engineer needs, which is experience.

A good engineer will have made hundreds of recordings using dozens of different microphones. Each session is an opportunity to make a new discovery. The engineer will make careful notes of the setup, and will listen to the results many times to build an association between the technique used and the sound achieved. Most of us do not have access to lots of professional microphones, but we could probably afford a pair of general purpose cardioids. With about \$400 worth of mics and a reliable tape deck, it is possible to learn to make excellent recordings. The trick is to record everything that will sit still and make noise, and study the results: learn to hear when the mic is placed badly and what to do about it. When you know all you can about your mics, buy a different pair and learn those. Occasionally, you will get the opportunity to borrow mics. If possible, set them up right alongside yours and make two recordings at once. It will not be long before you will know how to make consistently excellent recordings under most conditions.

Peter Elsea 1996

